King Fahd University of Petroleum & Minerals
Electrical Engineering Department
EE370: Communications Engineering I (101)

Major Exam II

December 18, 2010
6:45-08:15 PM
Building 59-2002

Serial #

0

Name: __________KEY________________

ID#______________________________

Question	Mark
1	/10
2	/11
3	/9
Total	/30

Instructions:

1. This is a closed-books/notes exam.
2. The duration of this exam is one and half hours.
3. Read the questions carefully. Plan which question to start with.
4. <u>CLEARLY LABEL ALL SIGNIFICANT VALUES ON BOTH AXIES OF ANY SKETCH.</u>
5. Work in your own. Keep your eyes on your paper.
6. Strictly no mobile phones are allowed.

<u>**Good luck**</u>

Dr. Ali H. Muqaibel

<u>Problem 1:</u> (10 points)

It is known that FM transmission requires more bandwidth than AM transmission. Mention three reasons for why would somebody use FM rather than AM. *(i.e Three advantages of FM over AM.) The three should be clear specific and distinct* **(1.5 points)**

1) Immunity to non-linearity

2) Constant Power

3) Trade-off bandwidth for signal quality (SNR)

For the following signal $x(t)$= 5.12 cos ($2\pi\, t + 2\, t^2 + \pi/4$), find the instantaneous frequency in Hz at t=2 sec

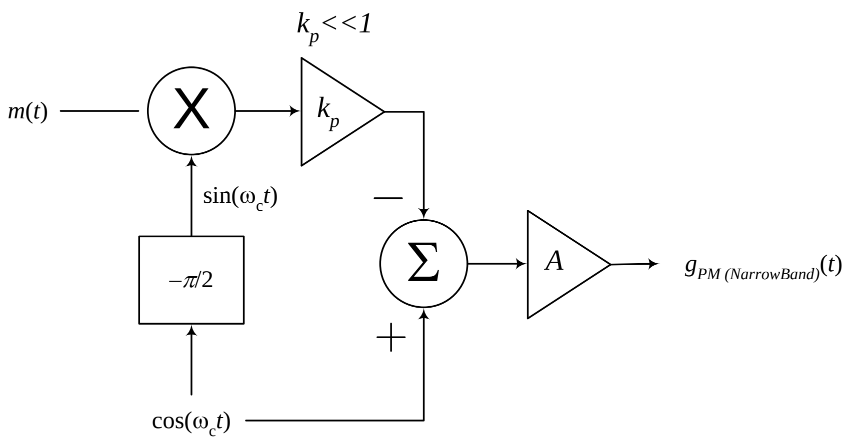
$\omega_i(t) = 2\pi + 4t \quad , \omega_i(t=2) = 2\pi + 4(2) = 2\pi + 8 \;, \qquad f_i = \frac{\omega_i}{2\pi} = 1 + \frac{8}{2\pi} = 2.2732$ Hz

(1.5 points)

Starting with $m(t)$ sketch the block diagram to get the narrowband phase modulated signal (NBPM)

(2 points)

$$\boxed{g_{PM\ (Narrowband)}(t) \approx A \cdot \left[\cos(\omega_c t) - k_p m(t) \sin(\omega_c t)\right]}$$



Narrowband PM Modulator

Mr. Ibraheem Al-Mawash has a Nokia phone (not Blackberry ;)). He wants to record a stream of 3 seconds.

For the video assume: 5 Mega-pixels/frame, 20 frames /sec, 3 colors, 8 bits/color/pixel.

For the sound assume: 16 bits/sample of sound using 22k sample/second.

a) How many bits are required to store the video +audio stream? **(2 points)**

For video= 5E6*20*3*8*3=7.2 Gbits

For audio=16*22k*3=1.056 Mbit

Total=7.201056 Gbits.

Note that data bits required for audio are much less than those for video.

b) How long does it take to send the above clip over bluetoot (assume 1 Mbit/sec is the transmission rate for Bluetooth) **(0.5 point)**

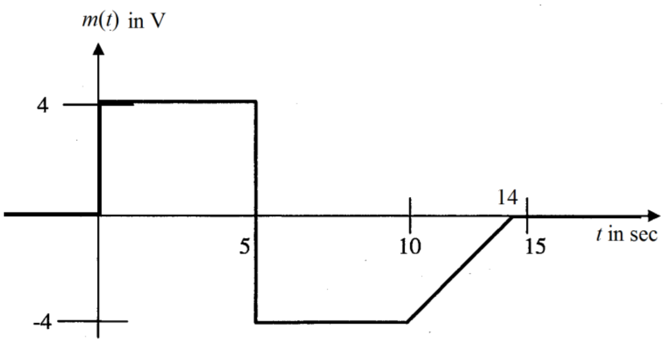
It takes 7.201056 Gbits./1Mbits/s= 7.2011e+003 sec= more than two hours! (need for compression)

The signal $s(t) = 8[1+\cos(120\pi t)\cos(100\pi t)]$ (t is in seconds) is sampled using an ideal sampling function (i.e. a periodic train of impulses) at the rate of 100 samples/sec. Each sample is quantized into the closest integer between 0 and 15. Each of the integer values is encoded using a 4-bit code word according to the usual binary representation of integers, (i.e. 0 = 0000, 1= 0001, …, 15 = 1111). Determine the sampled value, the quantized value and the binary code for the first three samples, starting from $t = 0$. **(2.5 points)**

time	0	0.01	0.02
Sampled value	16	14.4721	10.4721
Quantized value	15	14	10
Binary code	1111	1110	1010

Problem 2: **(11 points)**

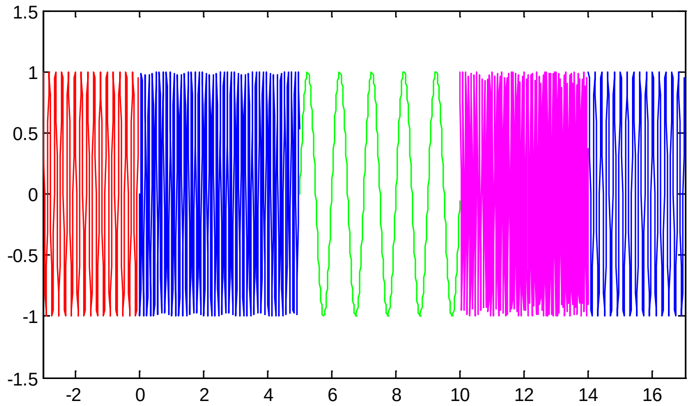
Consider the following message signal **(8 points)**



a) Plot the time domain signal for a frequency modulated signal when f_c=5Hz and k_f= 1 Hz/V

Show all important numbers on the sketch (phase or frequency). Sketch should be clear

$$f_i(t) = f_c + k_f m(t) = 5 + m(t)$$



Frequencies

5Hz

9Hz

1Hz

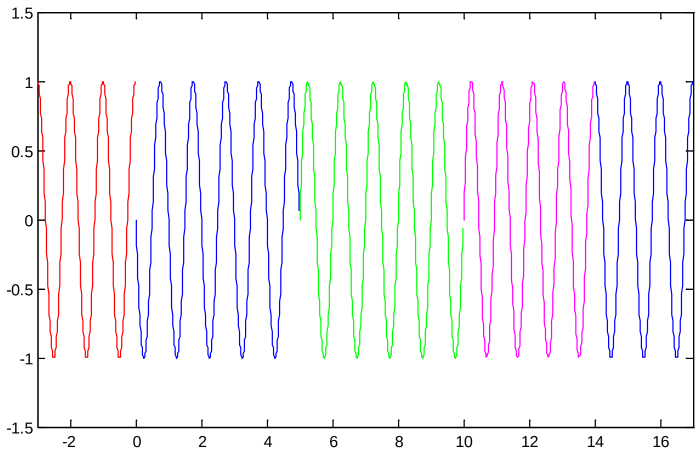
5+(-14+t)= -9+t (min 1 Hz , max 5Hz)

5Hz

b) Plot the time domain signal for a phase modulated signal when f_c=1 Hz and k_p= π/8 rad/V

$$g_{PM}(t) = A\cdot\cos\left[\omega_c t + k_p m(t)\right]$$

$$g_{PM}(t) = A\cdot\cos\left[2\pi(1)t + \frac{\pi}{8}m(t)\right]$$



A$\cos(2\pi t)$

-A$\sin(2\pi t)$

A$\sin(2\pi t)$

A$\cos(2\pi t-\pi/8(-14+t))$

A$\cos(2\pi t)$

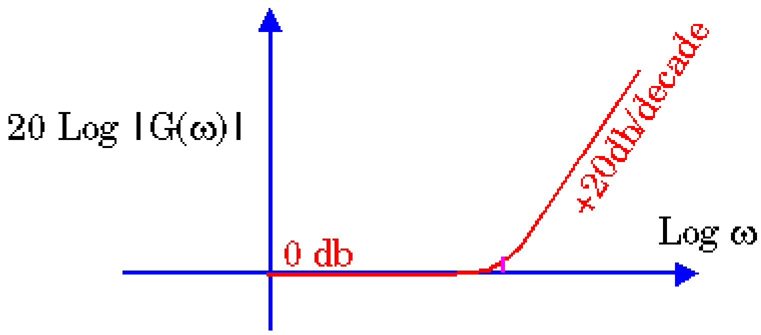
A satellite link sends a single NTSC-TV signal. The video signal is modulated onto the carrier using wideband frequency modulation, and the bandwidth of the transmitted modulated signal is 32 MHz. The baseband bandwidth of the TV signal is 4.2 MHz. Calculate the peak frequency deviation and the modulation index
(2 points)

Bandwidth RF=2(peak frequency deviation+ message bandwidth)
32M=2(Δf+4.2 M)
Δf =11.8 M Hz
The modulation index =$\Delta f / B$=2.8095

Sketch an example of a pre-emphasis filter (magnitude vs. frequency) and explain its idea **(1 point)**

Random FM system is proportional to frequency, with the effect that noise occurs predominantly at the highest frequencies within the baseband. This can be offset, to a limited extent, by boosting the high frequencies before transmission and reducing them by a corresponding amount in the receiver. Reducing the high frequencies in the receiver also reduces the high-frequency noise. These processes of boosting and then reducing certain frequencies are known as pre-emphasis and de-emphasis, respectively.

Any sketch that is fixed magnitude at low frequencies and the magnitude increases at high frequency.



Problem 3: **(9 points)**

Consider the following specifications for Mr.Al-Biladi Digital Telephony Company similar to the T1 system. In this system **10 voice channels** are converted to binary PCM and time-division multiplexed. Each telephone signal, with a nominal bandwidth of **3.4 kHz**, is sampled at a rate of **8000 samples/second.** The number of quantized levels is **128**. For the purpose of synchronization, **two bits** are inserted in the beginning of each frame.

(a) Sketch the structure of one frame. Show all significant time intervals. **(1 point)**

F 1	**F 2**	B1 1	B1 2	B1 3	B1 4	B1 5	B1 6	B1 7	B2 1	B2 2	B2 3	B2 4	B2 5	B2 6	B2 7	.	.	.	.	B10 1	B10 2	B10 3	B10 4	B10 5	B10 6	B10 7

Total of 2+7*10 =72 bits

(b) Determine the transmission rate of the system. **(2 points)**

Every frame contains 72 bits , time for every frame =1/8000=125E-6 sec.

Bit rate =72/125E-6 =72*8000=576 Kbits/ sec

(c) Compare the bandwidth required to transmit the 10 signals using the above Al-Biladi system to that required to transmit the same 10 signals in their analog form using DSB with FDM. **(1 point)**

[Assume the transmission bandwidth of a digital system =1/the bit duration=system transmission rate]

Using DSB for every user =2*3.4k=6.8kHz. For all users 6.8kHz*10=68kHz

For Digital Transmission Bandwidth= Data Rate= 576 kHz.

Digital Transmission (without compression) requires much more bandwidth compared with analog transmission

(d) If it is desired to reduce the transmission bandwidth of the digital system by simultaneously sampling at the minimum possible rate and reducing the number of bits per sample by one. Determine the new: **(3 points)**

(i) sampling rate (ii) number of quantized levels (iii) transmission bandwidth.

Sampling rate=2*3.4 Hz =**6.8 kHz**
Number of Quantization levels=2^6=**64**
New frame size=6*10=62 bits
Time for every frame=1/6.8k=147E-6 sec
Bit rate=62*6.8k=**421.6 kHz** (still much more than analog)

(e) For the original system in the question, the output SNR of the quantizer the 7-bit PCM was found to be 24 dB. The desired SNR is 42 dB. It was decided to increase the SNR to the desired value by increasing the number of quantization levels *L*. Find the fractional increase in the transmission bandwidth required for this increase in *L*. **(2 points)**
Adding 1 bit results in 6dB improvement. 42-24=18dB. So we need to add three bits per sample.
Number of bits per frame=2+(7+3)*10=102bits while the original number of bits was 72 bits
Since the BW is equivalent to the transmission rate. The increase in Bandwidth=102/72=1.4167. There is about 41.67% increase. Note that this is not equal to 3/7.

Dr. Ali Hussein Muqaibel